Performance of VoIP with DCCP for Satellite Links

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***Abstract*—We present experimental results for the performance of selected voice codecs using the Datagram Congestion Control Protocol (DCCP) with TCP-Friendly Rate Control (TFRC) congestion control mechanism over a satellite link. We evaluate the performance of both constant and variable data rate speech codecs (G.729, G.711 and Speex) for a number of simultaneous calls, using the ITU E-model and identify problem areas and potential for improvement. Our experiments are done on a commercial satellite service using a data stream generated by a VoIP application, configured with selected voice codecs and using the DCCP/CCID4 Linux implementation. We analyse the sources of packet losses which are a main contributor to reduced voice quality when using CCID4 and additionally analyse the effect of jitter which is one of the crucial parameters contributing to VoIP quality and has, to the best of our knowledge, not been considered previously in the published DCCP performance results. We propose modifications to the CCID4 algorithm and demonstrate how these improve the VoIP performance, without the need for additional link information other than what is already monitored by CCID4 (which is the case for Quick-Start). We also demonstrate the fairness of the proposed modifications to other flows. We identify the additional benefit of DCCP when used in VoIP admission control mechanisms and draw conclusions about the advantages and disadvantages of the proposed DCCP/ CCID4 congestion control mechanism for use with VoIP applications.**

I. Introduction

Voice over IP (VoIP) has become a well established technology with a large number of operators offering services and an ever growing number of end users. A large proportion of VoIP services use the public Internet, rather than a globally reserved bandwidth. This presents a problem both for the VoIP quality and the congestion of public Internet, as VoIP most commonly uses the UDP protocol which has no congestion control and no concept of fairness to other flows sharing the same network.

To bridge the gap between UDP and TCP, which is a reliable transport protocol and is not suitable for real time traffic, a new transport protocol for multimedia applications, Datagram Congestion Control Protocol (DCCP), has been proposed by IETF [1]. The main driver for having congestion control in an unreliable transport protocol was fairness to TCP traffic, which constitutes majority of the traffic on any Internet link. DCCP includes multiple congestion control algorithms identified by the Congestion Control ID (CCID), so that the application not needing reliable transport can select the appropriate congestion control method. CCID3 [2] and it's small packet variant CCID4 [3] rely on the TCP-Friendly Rate Control (TFRC) algorithm which is suitable for traffic with smooth changes in sending rates, such as telephony or video streaming. TFRC [4] is based on the TCP throughput equation and is therefore shown to be reasonably fair when competing with TCP flows. CCID3 is more suitable for streaming applications while CCID4 has been designed for applications with small packets like VoIP.

In geographically large countries with sparse population outside of the main centres like Australia, US or Canada and also in countries with a quickly growing infrastructure like India, there has been a number of new satellite network deployments in recent times [5],[6]. These networks have an increasing amount of multimedia and real time traffic and need to be considered in developing new protocols like DCCP.

In this paper, we present results of an experimental evaluation of the performance of selected voice codecs which use DCCP/CCID4 with TFRC congestion control over the IPSTAR satellite network [6] which is operational in Australia and a large number of countries in Asia. To the best of our knowledge, this is the first study presenting results from live satellite network performance measurements of VoIP using DCCP. We measure the receiver packet loss and delay and evaluate the VoIP quality under different conditions of network load using the ITU E-model [7]. We also evaluate fairness to competing TCP traffic sharing the same network. To mitigate the perceived packet loss resulting from DCCP/CCID4, we propose modifications to the CCID4 algorithm, which, compared to an alternative proposal Quick-Start [8], do not require any additional link rate information from the receiver. We demonstrate that the modifications result in a significantly improved VoIP quality compared to the original CCID4, while preserving the fairness advantage that CCID4 has over UDP.

The rest of the paper is structured as follows. Section II provides an overview of the related work and a description of the DCCP congestion control mechanism used for VoIP. Section III presents the experimental setup for live satellite tests. The following sections present experimental results, analysis of the results and our proposed modifications to the CCID4 protocol. In Section VI we present evaluation of the VoIP quality for different protocols. Section VII presents conclusions and a discussion of future work.

II. TFRC and Congestion Control for VoIP

This section presents an overview of the TFRC congestion control mechanism and a summary of related work.

DCCP/CCID3 [2] and DCCP/CCID4 [3] use TFRC [4] congestion control. In the TFRC congestion control mechanism, the appropriate sending rate is computed based on the monitored network conditions. Sender regulates the transmitted rate based on the received feedback messages which include the measured received rate, delay and an approximation of the packet loss rate. TFRC congestion control includes, similar to TCP, a slow start phase and a congestion avoidance phase.

In slow start, before the sender has received any receiver feedback, the sender's transmit rate X is set to one packet per second [2]. After the receiver feedback is available, the sender's initial rate is calculated as per equation (1):

$$X = \frac{min(4 \cdot s, max(2 \cdot s, 4380))}{RTT} \quad (1)$$

Where RTT is the estimated round trip time in seconds and s the packet mean size in bytes.

During the reminder of the slow start phase, the sender rate is increased with every received feedback, as per equation (2):

$$X = min(2 \cdot X, 2 \cdot X_{recv}) \quad (2)$$

Where X_{recv} is the receiver reported rate in bytes/second.

When the receiver reports the first error, TFRC enters the congestion avoidance phase, which uses equation (3) approximating the transmitted rate to what would be an equivalent rate of TCP under the same network conditions.

$$X = s \cdot f(p, RTT) \quad (3)$$

$$f(p, RTT) = \frac{1}{RTT \cdot \sqrt{\frac{p \cdot 2}{3}} + RTO \cdot \sqrt{\frac{p \cdot 27}{8}} \cdot p \cdot (1 + 32 \cdot p^2)}$$

Where:
p is the loss event rate.
RTO is the TCP retransmission timeout value in seconds.

CCID4 [3] differs from CCID3 only in the congestion avoidance phase. To calculate the sending rate X, in place of the packet size s in equation 3, CCID4 uses a fixed packet size of 1460 bytes modified by a header correction factor, according to the following equation (4):

$$X = 1460 \cdot \frac{s}{s + oh} \cdot f(p, RTT) \quad (4)$$

Where oh is the size of protocol overhead in bytes.

This ensures that the formula based rate is fair to both TCP and DCCP traffic, by using a common TCP packet size in place of the size of smaller VoIP packets.

In previous work, the performance of VoIP with DCCP/CCID4 protocol over satellite links has been studied in [9], [8] using simulation, and over generic links in [10] using emulation. The authors propose the use of Quick-Start [11] and Faster Restart [12] mechanisms and show that these methods provide only a partial improvement to the DCCP performance over a long delay network. In this paper, our intention was to analyse DCCP/CCID4 performance in a more dynamic environment than what has been considered in previous work and to provide additional insight into how DCCP handles real VoIP traffic.

In our previous work, we have proposed a dynamic computation of the number of sent DCCP/CCID3 feedback messages as a function of the end-to-end connection delay [8]. This modification greatly improves the rate computation of DCCP/CCID3 over long delay links by increasing the responsiveness of TFRC. The latter is achieved by a more accurate and timely estimation of network parameters. In this previous work, we aimed at achieving a data rate comparable to TCP when sending or receiving a high rate data stream using CCID3. In this paper, we push further the idea of dynamic adjustments, based on observed network conditions by investigating the parameters which will affect the perceived quality of VoIP carried by CCID4 over satellite links.

In the following section we present details of our experimental setup used to evaluate the VoIP performance with DCCP/CCID4.

III. EXPERIMENTAL SETUP

Our experimental setup at the NICTA Laboratory in Sydney, Australia is presented in Figure 1. We use the IPSTAR satellite service, with data being transmitted from the client side by the satellite modem, through the IPSTAR satellite gateway and the public Internet to our gateway (server side). For DCCP/CCID4, we use the experimental version of Linux kernel implementation, which we have modified to include our proposed changes as described in Section V.

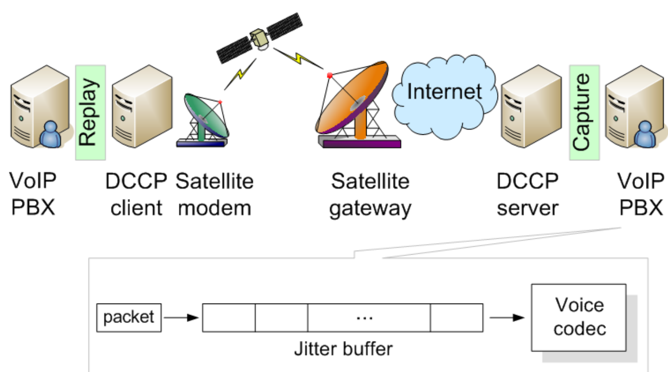

Figure 1. Experimental setup for live tests

The VoIP application used is Asterisk Private Branch Exchange (PBX) [13], with voice codecs commonly used in IP telephony. We use G.711 [14], Speex [15], with and without discontinuous transmission (DTX) and G.729 [16]. Table I lists details of the voice codecs used.

Table I
VOICE CODEC PARAMETERS

	voice frame size (bits)	voice frame size (msec)	data rate (kbits/s)
G.711	1440	10	64
G.729	160	10	8
Speex	variable	30	variable

To have a fair comparison of quality with different codecs and different transport protocols, we use a pre-recorded sample of speech which is one side of a 10 min conversation. The analogue wave file is played into the VoIP PBX, encoded with the appropriate voice codec, transmitted using UDP and the Inter Asterisk Exchange (IAX2) protocol [13] and captured at the receiving end. Our stream replicator application reads the UDP/IP payload and reproduces the timing and packet sizes of the VoIP packets. This data stream is transmitted using DCCP and captured at the receiving end for analysis. To produce examples of multiple conversations multiplexed into one data stream, we randomly start the pre-recorded conversation and apply the IAX2 multiplexing (trunking) option.

Default DCCP/CCID4 parameters are used in all simulations, including the sender buffer size of five packets, consistent with other published work.

Previous experimental results [17] characterised the IPSTAR satellite network, which we consider a good representative of the growing number of IP based satellite services. IPSTAR uses shared access over radio channels and consequently can have both congestion and errors on the link. The network RTT and loss characterised during our long term experiments show, for small packet sizes, an average RTT of around 1000msec and a packet error rate (PER) of 0.1%, consistent with published results which indicate an operating bit error rate (BER) of 10^{-7} [18]. Other satellite networks of interest, e.g. DVB-RCS [19], would have similar or lower PER for packets of the same size.

The following section presents a summary of tests on IPSTAR performed for the DCCP/CCID4 and the UDP protocol.

IV. TEST RESULTS AND PERFORMANCE ANALYSIS

We perform a number of experiments over the IPSTAR satellite network, for different voice codecs and under different load conditions. All experiments are of 10min duration. Groups of experiments were performed at the same time to minimise the impact of IPSTAR load conditions on test results.

We measure the packet loss rate, delay and jitter values at the DCCP receiver, which we will use to calculate the VoIP quality in Section VI. We also monitor parameters which contribute to the DCCP/CCID4 sender rate, including RTT, loss event rate p and receiver reported rate. A summary of the results from 10 IPSTAR experiments is presented in Table II.

Table II
AVERAGE PACKET ERROR RATE VALUES (%) FOR DIFFERENT CODECS MEASURED ON IPSTAR LINK

Voice Codec & load	Data rate (kbit/s)	CCID4 (%) (%)	UDP (%)
G.711	80	2.01	0.15
G.729	22	1.24	0.1
Speex	average 25	1.84	0.1
Speex/DTX	variable	17.3	0.1
Speex,5 calls	average 96	2	0.15
G.711,12 calls	780	6.32	1.55

It can be observed that the packet loss with CCID4 has significantly higher values than the packet loss for UDP.

Jitter values observed in the experiments are shown in Table III. Our observations show that the main source of jitter is the network rather than the DCCP congestion control algorithm, i.e. the average and maximum jitter values do not noticeably differ between DCCP and UDP experiments.

Table III
AVERAGE JITTER VALUES IN MILLISECONDS FOR ALL CODECS

Jitter (ms)	avg	max
UDP	54	1000
DCCP/CCID4	56	1000

The following section analyses the performance of DCCP in more detail and proposes an improvement to the algorithm.

A. Performance Analysis

To assist with analysis of the error rate results, we consider the potential sources of packet loss at the input of the voice codec decoder. These include:

1) packet loss between the application and the transport protocol, resulting from the inability of the transport protocol rate control to provide adequate sending rate to the application;

2) packet loss on the link, which can be due to errors and/or congestion (related to the DCCP error event rate p);

3) the loss resulting from jitter, as the voice codec will consider all packets which arrive with incorrect timing as lost.

It is important to note that the packet loss between the application and the transport protocol will only be applicable to DCCP, as UDP does not have a specific sender rate and it simply forwards application packets to the link.

The captured data for all experiments summarised in Table II indicates that only the experiments with G.711 had DCCP reported losses on the link and that all other losses were between the application and DCCP sender. Therefore, it can be concluded that the majority of losses are caused by the inability of DCCP/CCID4 protocol to provide a high enough sending rate to the VoIP application. Additionally, as there are no reported losses, CCID4 is operating in slow start phase and never reaches the (expected and desired) congestion avoidance phase in which the equation (4) ensures TCP fairness.

The following section presents our proposals to modify the DCCP/CCID4 protocol in a way which will enable better handling of the VoIP application traffic while bearing in mind the requirements for fairness to competing TCP flows.

V. IMPROVING DCCP/CCID4 FOR LONG DELAY LINKS

Experimental results indicate that significant VoIP packet losses occur in the slow start phase, when there is an initial transition from silence to speech and, if DTX is used, after other silence periods. Therefore, we propose to modify the CCID4 rate control in the following way.

In the first proposal, CCID4-N, we apply the existing CCID4 concept of replacing the measured packet size s by the equivalent packet size (of 1460 bytes modified by the header correction factor) to the slow start phase.

Consequently, in slow start, the sender's starting rate will remain one packet per second, but with the packet size modified according to our proposal. After the receiver feedback is available, the initial rate will be calculated by the following formula, which will replace equation (1) in rate calculations:

$$X = \frac{4380}{RTT} \cdot \frac{s}{s + oh} \qquad (5)$$

The proposed modification will increase the starting rate and the initial rate, which will result in less packet loss in transitions between silence and speech.

We also apply the CCID3 modification proposed in [17], so that N feedback messages per RTT are used by the receiver in place of the default one feedback per RTT, when RTT is longer than a nominal value of e.g. 100msec. This increases the speed of rate growth in slow start phase by applying formula 2 with increased frequency and, during the congestion control phase, provides more accurate values for changes of the RTT parameter used in formula 3 by more frequent measurements.

Our second proposal, CCID4-SCA, provides further optimisation for long delay links. We enhance the CCID4-N proposal by using a nominal value of RTT, e.g. 100msec in equation (5), which becomes:

$$X = 43800 \cdot \frac{s}{s + oh} \qquad (6)$$

The proposed modification will further increase the initial rate for links with a delay longer than 100msec, which should result in further reduction of packet loss in VoIP transmission in long delay links. As we only insert a “fixed” low RTT in calculating the initial rate, if the VoIP rate is too high for the potentially congested link, the standard CCID4 mechanism will detect and report errors and take DCCP into congestion avoidance phase.

A. *Performance Evaluation*

Table IV presents the summary of the packet error rate results from 10 IPSTAR experiments using the proposed CCID4 modifications, CCID4-N and CCID4-SCA. For comparison purposes we also include the CCID4 results from Table II.

Table IV
AVERAGE PACKET ERROR RATE VALUES (%) FOR DIFFERENT CODECS MEASURED IN EXPERIMENTS ON IPSTAR LINK

	CCID4 (%)	CCID4-N (%)	CCID4-SCA (%)
G.711	2.01	0.44	0.15
G.729	1.24	0.08	0.1
SPEEX	1.84	0.15	0.1
SPEEX/DTX	17.3	0.16	0.1
SPEEX/5calls	2	0.34	0.15
G.711/12calls	6.32	3.76	1.5

It can be observed that the packet error rate is significantly reduced by our proposals. CCID4-N improves the performance for lower rate VoIP streams and CCID4-SCA performs similarly to UDP. For higher number of calls producing data rates above the link rate, or in congested situations, our proposals will provide congestion control in the same way as CCID4.

In the following section we evaluate the VoIP quality for the different voice codecs, using the transport protocols and network load scenarios from our experiments.

VI. EVALUATION OF VOIP QUALITY

Mean Opinion Score (MOS) is an ITU defined quality metrics for voice [20]. As MOS is a subjective measurement which cannot be easily applied to a variety of changing network conditions, ITU has also defined an objective evaluation methodology, the E-Model [7], which enables evaluation of the voice quality based on measured network parameters. The E-Model's quality metrics is the R factor which can be used to calculate the MOS values as per [7].

We calculate the values of the R factor and MOS using the packet loss rate and measured delay. To calculate the packet loss rate resulting from jitter, we choose a buffer size of 400 msec, as a compromise between adding delay and loosing an increasing number of packets at the voice decoder. The jitter buffer size can be varied to further compensate for high jitter values, however that will not have an impact on the difference between the performance of DCCP and UDP as the measured jitter values for those protocols are very similar.

Speex is not an ITU codec and does not have the defined parameter values necessary for calculating the R factor. For the purpose of evaluating Speex quality, we roughly approximate the quality and error resilience of Speex codec to the corresponding parameters of the G.729 codec. We consider this approximation sufficiently accurate for the purpose of this evaluation, as the reported MOS score for the Speex codec used in our experiments is 4.1 [15], comparable to the MOS value of 4.18 for the G.729 codec, resulting from the E-Model calculations for the same network conditions.

Figure 2 shows the R factor for selected codecs and numbers of calls on the IPSTAR satellite network for all the transport protocols considered.

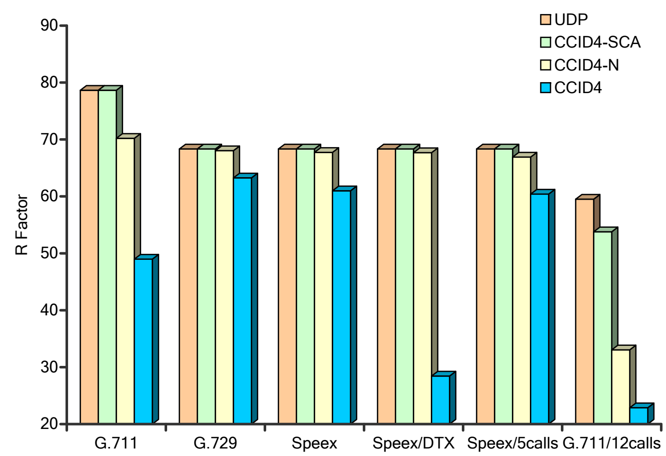

Figure 2. R factor for IPSTAR experiments, G.711, one and 12 calls, G.729, Speex, 1 and 5 calls, UDP, CCID4 and CCID4-N

To provide a clear view of the difference between the voice quality with UDP and with CCID4 variants, Table V presents the degradation in MOS values compared to UDP.

It can be observed that the R factor values on IPSTAR network range between an acceptable 79 (with a corresponding

Table V
DEGRADATION OF MOS VALUES FROM UDP FOR DIFFERENT CONGESTION CONTROL MECHANISMS ON THE IPSTAR NETWORK

	CCID4-N	CCID4	CCID4-SCA
G.711	0.36	1.45	0
G.729	0.01	0.25	0
SPEEX	0.03	0.37	0
SPEEX/DTX	0.03	1.97	0
SPEEX/5calls	0.07	0.40	0
G.711/12calls	1.34	1.73	0.3

MOS value of 3.9) for a G.711 call with either UDP or our proposal CCID4-SCA, to unacceptable values of below 50 for the same codec with CCID4 and even lower for Speex with DTX. Our proposals improve the voice quality compared to CCID4 for all cases considered and CCID4-SCA delivers voice quality similar to UDP for all but the highest number of calls.

In the following section, we will evaluate the fairness of the proposed DCCP/CCID4 modifications.

VII. FAIRNESS TO OTHER FLOWS

To evaluate fairness of multiple flows, we use Jain's fairness index [21].

As VoIP traffic is limited by the application, fairness can only be considered for VoIP streams which result from a number of parallel (multiplexed) calls which would require unfair capacity when sharing the link with other flows. We compare fairness to TCP of a VoIP data stream resulting from 12 parallel G.711 calls and additionally use a flow with rate equivalent to the nominal rate on the IPSTAR link. To illustrate the advantage of using DCCP, we also present the fairness results for UDP. The results of the fairness tests are presented in Table VI.

Table VI
FAIRNESS INDEX VALUES FOR TWO FLOWS, TCP AND CCID4 VERSIONS

Fairness Index	CCID4	CCID4-N	CCID4-SCA	UDP
G.711/12calls	0.9997	0.99997	0.99997	0.74
1Mbit/s data rate	0.985	0.98	0.98	0.5

It can be observed that both our proposals and CCID4 are equally fair to a TCP flow and that UDP, as expected, takes all the bandwidth it requires regardless of other flows.

The following section presents our conclusions and ideas for future work.

VIII. CONCLUSIONS AND FUTURE WORK

We have evaluated the performance of DCCP/CCID4 on a live satellite link for a number of scenarios which include different voice codecs and varying number of simultaneous VoIP calls. The main issue identified with using CCID4 for VoIP was in periods of transition from silence to speech, where in most cases CCID4 cannot support the required application rate and produces significant packet losses, particularly on long delay links. We have proposed modifications to CCID4 which mitigate this problem by enabling a faster slow start, higher minimum rate and a more accurate parameter measurement resulting in a more responsive rate adjustment which better matches the varying network conditions. Both our proposals require minimal changes to CCID4 specification and no interworking with other network components. They result in no loss of fairness to TCP traffic compared to the original CCID4.

We believe that with the proposed improvements and the fairness it was designed for, CCID4 has a significant advantage over UDP. We would also like to point out another benefit which DCCP can provide over UDP for VoIP: DCCP awareness of transport layer losses can be used for VoIP call admission control. As the number of calls increases and reaches the level where packets are lost, the DCCP measured loss rate can be used to trigger blocking of new calls by the VoIP application.

In continuation of this work, we plan to further investigate DCCP aided call admission control and codec rate management for VoIP.

IX. ACKNOWLEDGEMENT

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